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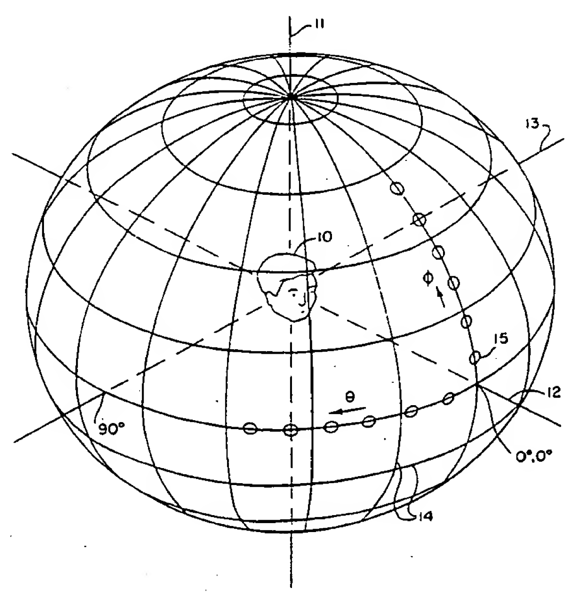
(54) Title: METHODS AND APPARATUS FOR PRODUCING DIRECTIONAL SOUND

(57) Abstract

Free-field-to-eardrum transfer functions (FETF's) are developed by comparing auditory data for points in three-dimensional space for a model ear and auditory data collected for the same listening location with a microphone. Each FETF is represented as a weighted sum of frequency-dependent functions obtained from an expansion of the measured FETF's covariance matrix. Spatial transformation characteristic functions (STCF's) are applied to transform the weighted frequency-dependent factors to functions of spatial variables for azimuth and elevation. A generalized spline model is fit to each STCF to filter out noise and permit interpolation of the STCF between measured points. Sound is reproduced for a selected direction by synthesizing the weighted frequency-dependent factors with the smoothed and interpolated STCF's.



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METHODS AND APPARATUS FOR PRODUCING DIRECTIONAL SOUND

Background of the Invention

1. Field of the Invention

The field of the invention is methods and apparatus for detecting and reproducing sound.

2. Description of the Background Art

Extensive physical and behavioral studies have revealed that the external ear (including torso, head, pinna, and canal) plays an important role in spatial hearing. It is known that external ear modifies the spectrum of incoming sound according to incidence angle of that sound. It is further known that in the context of binaural hearing, the spectral difference created by the external ears introduces important cues for localizing sounds in addition to interaural time and intensity differences. When the sound source is within the sagittal plane, or in the case of monaural hearing, the spectral cues provided by the external ear are utilized almost exclusively by the auditory system to identify the location of the sound source. The external ears also externalize the sound image. Sounds presented binaurally with the original time and intensity differences but without the spectral cues introduced by the external ear are typically perceived as originating inside the listener's head.

Functional models of the external ear transformation characteristics are of great interest for simulating realistic auditory images over headphones. The problem of reproducing sound as it would be heard in three-dimensional space occurs in hearing research, high fidelity music reproduction, and voice communication.

Kistler and Wightman describe a methodology based on

free-field-to-eardrum transfer functions (FETF's) in a paper published in the Journal of the Acoustical Society of America (March, 1992) pp. 1637-1647. This methodology analyzes the amplitude spectrum and the results represent up to 90% of the energy in the measured FETF amplitude. This methodology does not provide for interpolation of the FETF's between measured points in the spherical auditory space around the listener's head, or represent the FETF phase.

For further background art in the relevant area of auditory research, reference is made to the Introduction portion of our article, "External Ear Transfer Function Modeling: A Beamforming Approach", published in the Journal of the Acoustical Society of America, vol. 92, no. 4, Pt. 1 (October 30, 1992) pages 1933-1944.

Summary of the Invention

The invention is incorporated in methods and apparatus for recording and playback of sound, and sound recordings, in which a non-directional sound is processed for hearing as a directional sound

Using measured data, a model of the external ear transfer function is derived, in which frequency dependance is separated from spatial dependance. A plurality of frequency-dependent functions are weighted and summed to represent the external ear transfer function. The weights are made a function of direction. Sounds that carry no directional cues are perceived as though they are coming from a specific direction when processed according to the signal processing techniques disclosed and claimed herein.

With the invention, auditory information takes on a spatial three-dimensional character. The methods and apparatus of the invention can be applied when a listener, such as a pilot, astronaut or sonar operator needs direc-

tional information, or they can be used to enhance the pleasurable effects of listening to recorded music.

Other objects and advantages, besides those discussed above, shall be apparent to those of ordinary skill in the art from the description of the preferred embodiment which follows. In the description, reference is made to the accompanying drawings, which form a part hereof, and which illustrate examples of the invention. Such examples, however, are not exhaustive of the various embodiments of the invention, and therefore reference is made to the claims which follow the description for determining the scope of the invention.

Brief Description of the Drawings

Fig. 1 is a diagram showing how sound data is collected according to the present invention;

Figs. 2a-2j are spectral graphs of sound collected in Fig. 1 or interpolated relative to data collected in Fig. 1;

Fig. 3 is a block diagram of the apparatus used to record sound data as depicted in Figs. 1 and 2;

Fig. 4 is a flow chart showing the steps in producing a sound according to the present invention;

Fig. 5a is a functional circuit diagram showing how a directional sound is synthesized with the apparatus of Fig. 6;

Fig. 5b is a functional circuit diagram showing a second method for synthesizing sound with the apparatus of Fig. 6; and

Fig. 6 is a block diagram showing apparatus for producing a directional sound according to the present invention.

Detailed Description of the Preferred Embodiments

Referring to Fig. 1, the invention utilizes data measured in three-dimensional space relative to a typical human ear. The measurements may be conducted on a human subject, if a specific subject ear is required, or with a special manikin head 10, such as a KEMAR™ head, which represents a typical human ear. The spherical space around the head is described in terms of spherical coordinates θ and φ. The variable θ represents azimuth angle readings relative to a vertical midline plane defined by axes 11 and 12 between the two ears (with angles to the right of the midline plane in Fig. 1 being positive angles and with angles to the left being negative angles). The variable φ represents elevation readings relative to a horizontal plane passing through the axes 12 and 13 and the center of the ears (above this plane being a positive angle and below this plane being a negative angle). Isoazimuth and isoelevation lines 14 are shown in 20° increments in Fig. 1. A speaker 15 is moved to various positions and generates a broadband sound.

The ear sound is measured using the subject's ear or manikin's head 10 by placing a microphone in one ear to record sound as it would be heard by a listener. Data can be taken for both ears. To develop a free-field-to-ear transfer function, sound is also measured without the effects of the ear, by removing the subject's ear or manikin's head 10 and detecting sound at the ear's previous location. This is "free field" sound data. Both measurements are repeated for various speaker locations. Standard signal processing methods are used to determine the transfer function between the ear and the free-field data at each location.

Figs. 2a, 2c, 2e, 2g and 2i shows a series of spectral sound graphs (amplitude vs. frequency) for a series

of readings for 18.5° elevation, and varying azimuth angles from 0° to 36°. The readings were taken at 9° intervals. A shift in spectral peaks and valleys is observed as the origin of the sound is moved. Figs. 2b, 2d, 2f, 2h and 2j show values which have been interpolated using the data and methodology described herein.

Fig. 3 illustrates the apparatus for collecting sound data for free-field and ear canal recording. The subject 10 and a movable speaker 15 are placed in a chamber 16 for sound recording. A personal computer 20, such as the IBM PC AT or an AT-compatible computer, includes a bulk memory 21, such as a CD-ROM or one or more large capacity hard drives. Microphones 23a, 23b are placed in the subject's or manikin's ears. The sound is processed through an amplifier and equalizer unit 24 external to the computer 20 and analog band pass filtering circuitry 27 to an A-to-D converter portion 22a of a signal processing board in the computer chassis. There, the analog signals of the type seen in Fig. 2 are converted to a plurality of sampled, digitized readings. Readings are taken at as many as 2000 or more locations on the sphere around the manikin head 10. This may require data storage capacity on the order of 70 Megabytes.

The computer 20 generates the test sound through a sound generator portion 22b of the signal processing board. The electrical signal is processed through power amplifier circuitry 25 and attenuator circuitry 26 to raise the generated sound to the proper power level. The sound-generating signal, which is typically a square wave pulse of 30-100 microseconds in duration or other broadband signal in duration is then applied through the speaker 15 to generate the test sound. The speaker 15 is moved from point to point as shown in Fig. 1.

In an alternative embodiment for recording spatial sound data, a VAX 3200 computer is used with an ADQ-32 signal processing board.

In methods and apparatus for recording and playing back simulated directional sound, an audio input signal is passed through a filter whose frequency response models the free field-to-eardrum transfer function. This filter is obtained as a weighted combination of basic filters where the weights are a function of the selected spatial direction.

Fig. 4 illustrates how sound data collected in Figs. 1-3 is processed to determine the basic filters and weights used to impart spatial characteristics to sound according to the present invention. The sound data has been input and stored for a plurality of specific speaker locations, as many as 2000 or more, for both free field, $R(\omega, \theta, \varphi)$, and ear canal recording, $E(\omega, \theta, \varphi)$. This is represented by input block 31 in Fig. 4. This data typically contains noise, measurement errors and artifacts from the detection of sound. Conventional, known signal processing techniques are used to develop a free-field-to-ear transfer function $H(\omega, \theta, \varphi)$, as represented by process block 32 in Fig. 4, which is a function of frequency ω, at some azimuth θ and some elevation φ. This block 32 is executed by a program written in MATLAB and C programming language running on a SUN/SPARC 2 computer. MATLAB™, version 3.5, is available from the Math Works, Inc., Natick, MA. A similar program could be written for the AT-compatible computer 20 or other computers to execute this block.

If $H(\omega, \theta, \varphi)$ is the measured FETF at some azimuth θ and elevation φ, the overall model response, $\hat{H}(\omega, \theta, \varphi)$, can be expressed as the following equation:

$$\hat{H}(\omega, \theta, \varphi) = \sum_{i=1}^{p} t_i(\omega)\, w_i(\theta, \varphi) + t_0(\omega) \qquad (1)$$

Note that the model separates frequency-dependence characterized by the basic filters, represented by $t_i(\omega)$ (i = 0,

$1,\ldots, p)$, also referred to as eigenfilters (EF's), from the spatial-dependence represented by weights, $w_i(\theta, \varphi)$ ($i = 1,\ldots,p$). These weights are termed spatial transformation characteristic functions (STCF's). This provides a two-step procedure for determining $\hat{H}(\omega, \theta, \varphi)$ for live recordings provided that the above equation can be solved for $t_i(\omega)$ and $w_i(\theta, \varphi)$.

The present invention provides the methods and apparatus to determine EF's and STCF's, so that the model response $\hat{H}(\omega, \theta, \varphi)$ is a good approximation to $H(\omega, \theta, \varphi)$.

Let $\mathbf{H}(\theta, \varphi)$ and $\mathbf{t}_i$ be N dimensional vectors whose elements are N samples in frequency of the measured FETF's, $H(\omega, \theta, \varphi)$, and N samples in frequency of the eigenfilters $\{\mathbf{t}_i(\omega), i = 0,1,\ldots,p\}$. The value for N is typically 256 although larger or smaller values could also be used. N should be sufficiently large so that the eigenfilters are well described by the samples of $\mathbf{t}_i$. The eigenfilters $\{\mathbf{t}_i(\omega), i = 1,2,\ldots,p\}$ are chosen as eigenvectors corresponding to the p largest eigenvalues of a sample covariance matrix Σ_H formed from the spatial samples of the FETF frequency vectors $\mathbf{H}(\theta, \varphi)$. The eigenfilter $\mathbf{t}_0(\omega)$ is chosen as the sample mean $\overline{\mathbf{H}}$ formed from the spatial samples of FETF frequency vectors $\mathbf{H}(\theta, \varphi)$. If $\mathbf{H}(\theta_j, \varphi_k)$ represents the measured FETF at the azimuth elevation pair (θ_j, φ_k) and providing that $j = 1,\ldots, L$, $k = 1,\ldots, M$, where LxM is on the order of 2000, the covariance matrix Σ_H of FETF samples is given by

$$\Sigma_H = \frac{1}{LM} \Sigma_{j=1}^{L} \Sigma_{k=1}^{M} \alpha_{jk} (\mathbf{H}(\theta_j, \varphi_k) - \overline{\mathbf{H}}) (\mathbf{H}(\theta_j, \varphi_k) - \overline{\mathbf{H}})^H \quad (2)$$

where $\overline{\mathbf{H}}$, the sample mean, is expressed as follows:

$$\overline{\mathbf{H}} = \frac{1}{LM} \Sigma_{j=1}^{L} \Sigma_{k=1}^{M} \mathbf{H}(\theta_j, \varphi_k) \quad (3)$$

In equation (2) the superscript "H" denotes a complex

conjugate transpose operation. The non-negative weighting factor α_{jk} is used to emphasize the relative accuracy of this analysis in some directions more than others. If all directions are equally important, $\alpha_{jk} = 1$, for $j = 1,\ldots, L$, $k = 1,\ldots, M$.

The EF frequency vectors $\{\mathbf{t}_i(\omega)\ (i = 1, 2,\ldots, p)\}$ satisfy the following eigenvalue problem

$$\Sigma_H\ \mathbf{t}_i = \lambda_i\ \mathbf{t}_i \qquad (4)$$

where $i = 1, \ldots, p$ and where λ_i are the "p" largest eigenvalues of Σ_H. The fidelity of sound reproduced using the methodology of the invention is improved by increasing "p". A typical value for "p" is 16. The EF vector, $\mathbf{t}_0(\omega)$ is set equal to $\overline{\mathbf{H}}$.

The STCF's $\mathbf{w}_i(\theta,\varphi)$, $i = 1,\ldots,p$, are obtained by fitting a spline function over azimuth and elevation variables to STCF samples, $\tilde{\mathbf{w}}_i(\theta_j,\varphi_k)$, $i = 1,\ldots,p$, $j = 1, \ldots, L$, $k = 1, \ldots, M$, which are chosen to minimize the squared error between the calculated values and measured values of FETF's at locations (θ_j,φ_k) $j = 1, \ldots, L$, $k = 1, \ldots, M$. The samples $\tilde{\mathbf{w}}_i(\theta_j,\varphi_k)$ that minimize the squared error are given by

$$\tilde{\mathbf{w}}_i(\theta_j,\varphi_k) = \mathbf{t}_i^H\ \mathbf{H}(\theta_j,\varphi_k) \qquad (5)$$

where $i = 1,\ldots,p$, $j = 1, \ldots, N$, $k = 1, \ldots, M$. Here we assume without loss of generality that the $\mathbf{t}_i$ has a unit norm, that is, $\mathbf{t}_i^H\ \mathbf{t}_i = 1$, $i = 1,\ldots,p$.

The spline model for generating the STCF's smooths measurement noise and enables interpolation of the STCF's (and hence the FETF's) between measurement directions. The spline model is obtained by solving the regularization problem

$$\min_{w_i(\theta,\varphi)} \sum_{j=1}^{L}\sum_{k=1}^{M} \left(\tilde{w}_i(\theta_j,\varphi_k) - w_i(\theta_j,\varphi_k)\right)^2 + \lambda\,|\,Pw_i(\theta,\varphi)\,|^2 \qquad (6)$$

where i = 1, ...,p. Here $w_i(\theta_j,\varphi_k)$ is the functional representation of the *ith* STCF, λ is the regularization parameter, and P is a smoothing operator.

The regularization parameter controls the trade-off between the smoothness of the solution and its fidelity to the data. The optimal value of λ is determined by the method of generalization cross validation. Viewing θ and φ as coordinates in a two dimensional rectangular coordinate system, the smoothing operator P is

$$|Pw_i(\theta,\varphi)|^2 = \int_R d\theta d\varphi \left\{ \left[\frac{\partial^2 w(\theta,\varphi)}{\partial\theta^2}\right]^2 + 2\left[\frac{\partial^2 w(\theta,\varphi)}{\partial\theta\,\partial\varphi}\right]^2 + \left[\frac{\partial^2 w(\theta,\varphi)}{\partial\varphi^2}\right]^2 \right\} \quad (7)$$

The regularized STCF's are combined with the EF's to synthesize regularized FETF's at any given θ and φ.

Process block 33 in Fig. 4 represents the calculation of Σ_H, which is performed by a program in the MATLAB™ language running on the SUN/SPARC 2 computer. A similar program could be written for the AT-compatible computer 20 or another computer to execute this block.

Next, as represented by process block 34 in Fig. 4, an eigenvector expansion is applied to the Σ_H results to calculate the eigenvalues, λ_i, and eigenvectors $\mathbf{t}_i$ which correspond to the weighted functions of frequency $t_i(\omega)$. In this example, the eigenanalysis is more specifically referred to as the Karhunen-Loeve expansion. For further explanation of this expansion, reference is made to Papoulis, Probability, Random Variables and Stochastic Processes, 3d ed. McGraw-Hill, Inc., New York, NY, 1991, pp. 413 – 416, 425. The eigenvectors, are then processed, as represented by block 35 in Fig. 4, to calculate the samples of the STCF's, $\mathbf{w}_i$ as a function of spatial variables (θ, φ) for each direction from which the sound has been measured, as described in equation 5 above. This calculation is performed by a program in the MATLAB™ language

running on the SUN/SPARC computer. A similar program could be written for the AT-compatible computer 20 or a different computer to execute this block.

Next, as represented by process block 36 in Fig. 4, a generalized spline model is fit to the STCF samples using a publicly available software package known as RKpack, obtained through E-mail at netlib@Research.att.com.. The spline model filters out noise from each of the sampled STCF's. The spline-based STCF's are now continuous functions of the spatial variables (θ, φ).

The surface mapping and filtering provides resulting data which permits interpolation of the STCF's between measured points in spherical space. The EF's $\mathbf{t}_0(\omega)$ and $\mathbf{t}_i(\omega)$, and the STCF's, $\mathbf{w}_i(\theta, \varphi)$, i=1, ..., p, describe the completed FETF model as represented in process block 37. An FETF for a selected direction is then synthesized by weighting and summing the EF's with the smoothed and interpolated STCF's. A directional sound is synthesized by filtering a non-directional sound with the FETF as represented by process block 38.

The synthesized sound is converted to an audio signal, as represented by process block 39, and converted to sound through a speaker, as represented by output block 40. This completes the method as represented by block 41.

Fig. 5a is a block diagram showing how a directional sound is synthesized according to the present invention. A non-directional sound represented by input signal 29 in Fig. 5 is played back through a variable number, p, of filters 42 corresponding to a variable number, p, of EF's for the right ear and a variable number, p, of filters 43 for the left ear. In this embodiment p = 16 is assumed for illustrative purposes. The signal coming through each of these sixteen filters 42 is amplified according to the SCTF analysis of data, represented by blocks 106, 107 as a function of spatial variables θ and φ, as outlined above, for each ear as represented by sixteen multiplying junc-

tions 74 for the right ear and sixteen multiplying junctions 75 for the left ear. The input signal 29 is also filtered by the FETF sample mean value, $t_0(\omega)$, represented by blocks 51, 52 in Fig. 5a, and then amplified by a factor of unity (1). The amplified and EF filtered component signals are then summed with each other and with the zero-frequency components 51, 52 at summing junctions 80 and 81, for right and left ears, respectively, and played back through headphones to a listener in a remote location. By weighting the EF filtered signals with the STCF weights corresponding to a selected direction defined by θ and φ, and summing the weighted filtered signals, a sound was produced with the effect that the sound was originating from the selected direction.

Fig. 5b shows an alternative approach to synthesize directional sound according to the present invention. Here the non-directional input signal 29 is filtered directly by the FETF for the selected direction. The FETF for the selected direction is obtained by weighting the EF's 55, 56 at "p" multiplying junctions 45, 46 with the STCF's 106, 107 for the selected direction. Then, the adjusted EF's are summed at summing junctions 47, 48, together with the FETF sample mean value, $t_0(\omega)$, represented by elements 55, 56, to provide a single filter 49, 50 for each respective ear with a response characteristic for the selected direction of the sound.

In the above examples, the filtering of components is performed in the frequency domain, but it should be apparent that equivalent examples could be set up to filter components in the time domain, without departing from the scope of the invention.

Both Figs. 5a and 5b show a final stage which accounts for the interaural time delay. Since the interaural time delay was removed during the process of the modeling, it needs to be restored in the binaural implementation. The interaural time dely ranges from 0 to about

700μs. The blocks 132 and 142 in Figs. 5a and 5b, respectively, represent interaural time delay controllers. They convert the given location variables θ and φ into time delay control signals and send these control signals to both ear channels. The blocks 130, 131, 140 and 141 are delays controlled by the interaural time delay controllers 132, 142. The actual interaural time delay can be calculated by cross-correlating the two ear canal recordings vs. each sound source location. These discrete interaural time delay samples are then input into the spline model, thus a continuous interaural time delay function is acquired.

Fig. 6 is a block diagram showing apparatus for producing the directional sound according to the present invention. The non-directional sound is recorded using a microphone 82 to detect the sound and an amplifier 83 and signal processing board 84-86 to digitize and record the sound. The signal processing board includes data acquisition circuitry 84, including analog-to-digital converters, a digital signal processor 85, and digital-to-analog output circuitry 86. The signal processor 85 and other sections 84, 86 are interfaced to the PC AT computer 20 or equivalent computer as described earlier. The digital-to-analog output circuitry 86 is connected to a stereo amplifier 87 and stereo headphones 88. The measured data for the FETF is stored in mass storage (not shown) associated with the computer 20. Element 89 illustrates an alternative in which an audio signal is prerecorded, stored and then fed to the digital signal processor 85 for production of directional sound.

The signal 29 in Figs. 5a and 5b is received through microphone 82. The filtering by filters 42 and 43, and other operations seen in Fig. 5a and 5b, are performed in the digital signal processor 85 using EF's and STCF function data 106, 107 received from the AT-compatible computer 20 or other suitable computer.

The other elements 86-88 in Fig. 6 convert the audio

signals seen Fig. 5 to sound which the listener observes as originating from the direction determined by selection of θ and φ in Fig. 5. That selection is carried out with the AT- compatible computer 20, or other suitable computer, by inputting data for θ and φ.

It should be apparent that this method can be used to make sound recordings in various media such as CD's, tapes and digitized sound recordings, in which non-directional sounds are converted to directional sounds by inputting various sets of values for θ and φ. With a series of varying values, the sound can be made to "move" relative to the listener's ears, hence, the terms "three-dimensional" sound and "virtual auditory environment" are applied to describe this effect.

This description has been by way of example of how the invention can be carried out. Those of ordinary skill in the art will recognize that various details may be modified in arriving at other detailed embodiments, and that many of these embodiments will come within the scope of the invention. Therefore to apprise the public of the scope of the invention and the embodiments covered by the invention the following claims are made.

CLAIMS

We claim:

1. A method of sound playback in which a sound is to be produced as if originating from a selected direction in space relative to a listener's ear, the method comprising the steps of:

storing a recorded audio signal representing a sound to be played back;

converting the recorded audio signal to at least one filtered component;

adjusting the filtered component as a function of the selected direction in space from which the direction of the sound is to be simulated upon play back;

generating a resultant audio signal in response to the adjusted, filtered component;

converting the audio signal to a sound observed by a listener to have originated at the selected direction in space relative to the listener's ear.

2. The method of claim 1, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

3. The method of claim 1, wherein

during the converting step the recorded audio signal is converted to a plurality of filtered components;

wherein the adjusting step is performed for the plurality of filtered components; and

further comprising the step of summing the resulting adjusted filtered components and generating a resultant audio signal.

4. The method of claim 3, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

5. Apparatus for playback of sound, wherein a sound is to be produced as if originating at a selected direction in space relative to a listener's ear, the apparatus comprising:

means for storing an audio signal representing a sound to be played back;

means for converting the audio signal to a plurality of filtered components based on premeasured data for a sound detected within an ear;

means for adjusting each filtered component as a function of the selected direction in space from which the direction of the sound is to be simulated upon playback;

means for summing the resulting adjusted, filtered components of the sound to be played back and generating a resultant audio signal;

means for converting the audio signal to a sound observed by a listener to have originated at the selected direction in space relative to the listener's ear.

6. A method for recording a sound to be played back as if originating at a selected direction in space relative to a listener's ear, the method comprising:

receiving and storing an audio signal representing a sound to be played back;

converting the recorded audio signal to at least one filtered component;

adjusting the filtered component as a function of the selected direction in space from which the direction of the sound is to be simulated upon play back; and

storing the the adjusted filtered component in a recording medium for playback.

7. The method of claim 6, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

8. The method of claim 6, wherein

during the converting step the recorded audio signal is converted to a plurality of filtered components;

wherein the adjusting step is performed for the plurality of filtered components; and

further comprising the step of summing the resulting adjusted filtered components and generating a resultant audio signal.

9. The method of claim 8, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

10. A sound recording made according to a method including the steps of:

receiving and storing an audio signal representing a sound to be played back;

converting the recorded audio signal to at least one filtered component;

adjusting the filtered component as a function of the selected direction in space from which the direction of the sound is to be simulated upon play back; and

storing the the adjusted filtered component in a recording medium for playback.

11. The method of claim 10, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

12. The method of claim 10, wherein

during the converting step the recorded audio signal is converted to a plurality of filtered components;

wherein the adjusting step is performed for the plurality of filtered components; and

further comprising the step of summing the resulting adjusted filtered components and generating a resultant audio signal.

13. The method of claim 12, wherein

the adjusting step is performed in response to premeasured data for a plurality of directions in space; and

and further comprising the step of applying a spline function to the premeasured data,

and the step of interpolating results from the spline function operation corresponding to directions in space which are in between directions in space for the premeasured data.

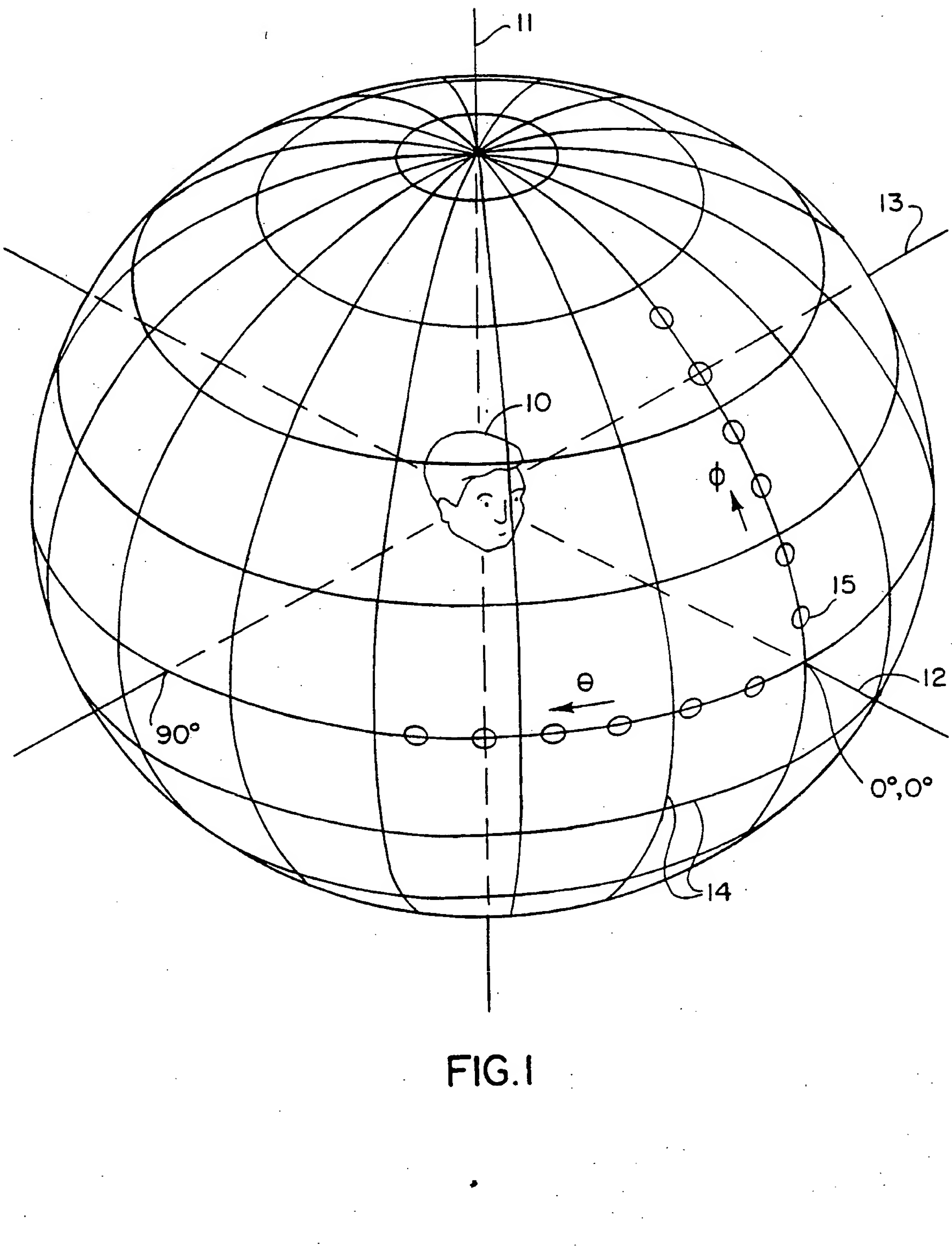


FIG.1

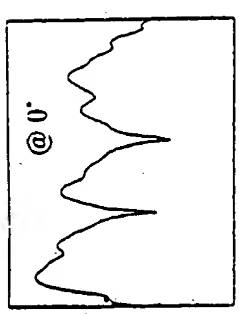


Fig. 2a

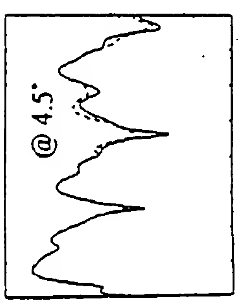


Fig. 2b

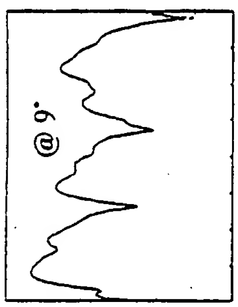


Fig. 2c

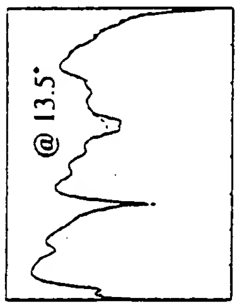


Fig. 2d

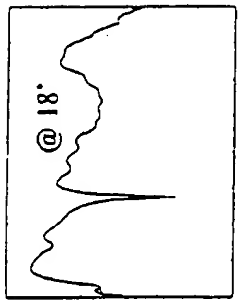


Fig. 2e

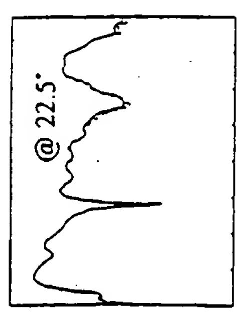


Fig. 2f

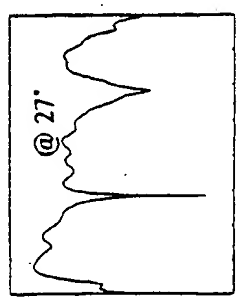


Fig. 2g

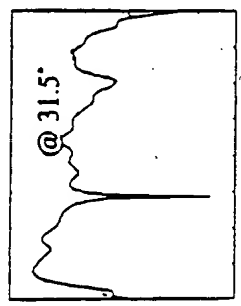


Fig. 2h

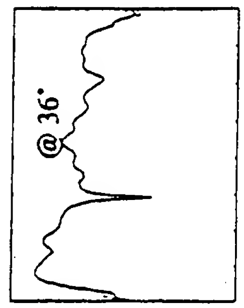


Fig. 2i

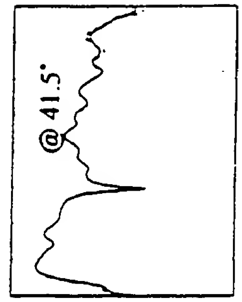


Fig. 2j

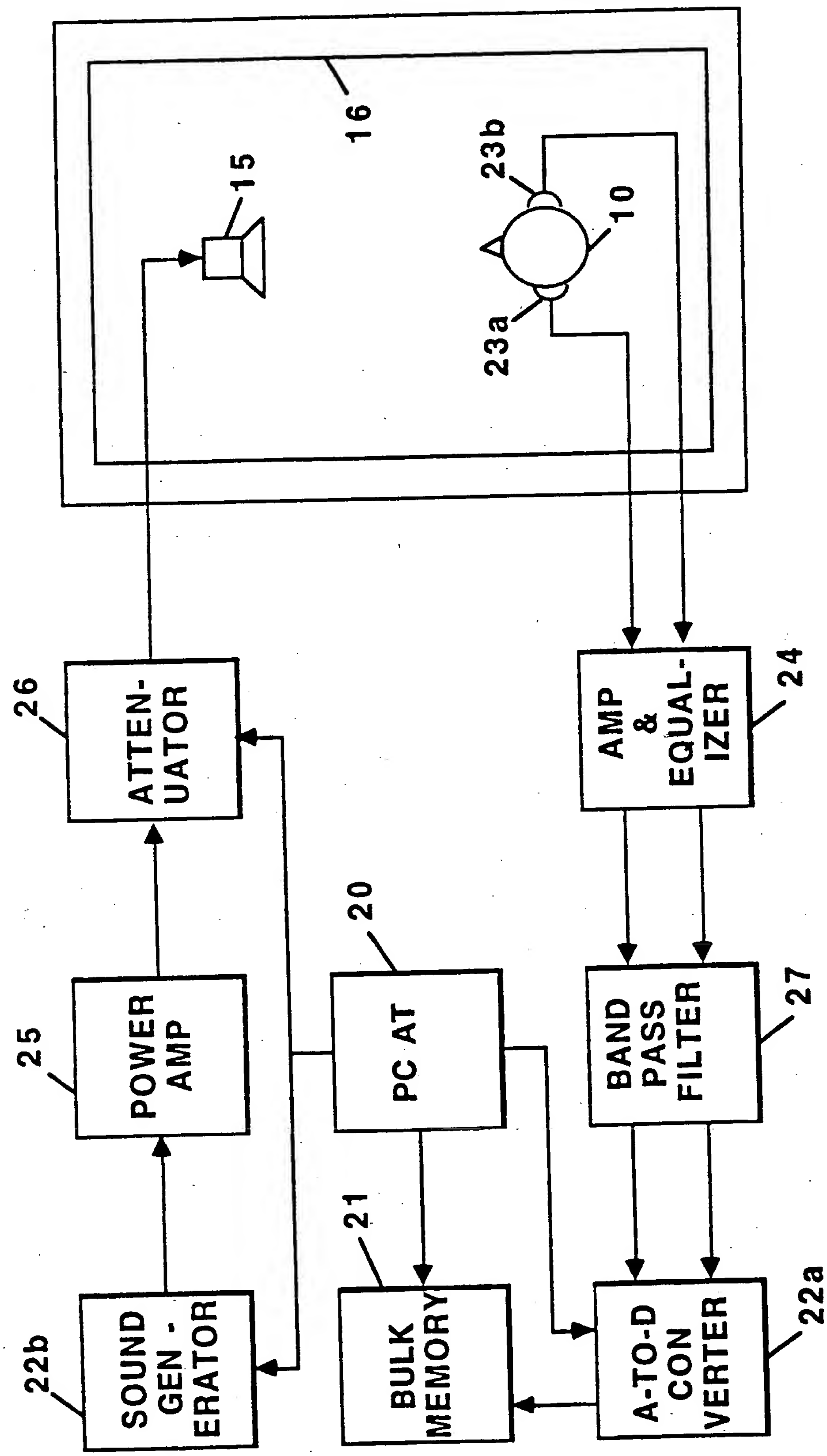

16
15
23b
10
23a
26
ATTEN-UATOR
24
AMP & EQUAL-IZER
25
POWER AMP
20
PC AT
27
BAND PASS FILTER
22b
SOUND GEN - ERATOR
21
BULK MEMORY
22a
A-TO-D CON VERTER

Fig. 3

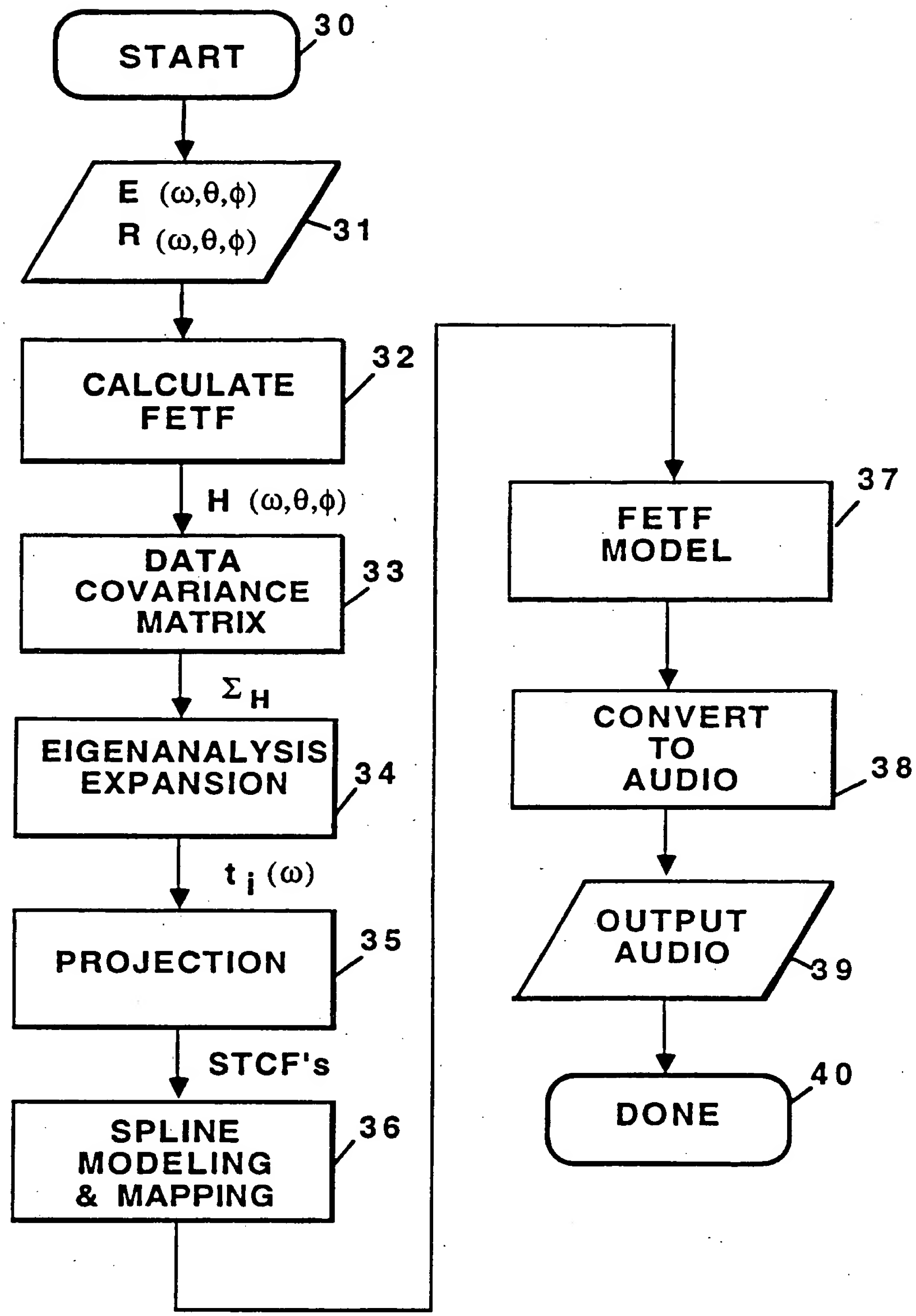

START
30
E (ω,θ,ϕ)
R (ω,θ,ϕ)
31
CALCULATE FETF
32
H (ω,θ,ϕ)
DATA COVARIANCE MATRIX
33
Σ H
EIGENANALYSIS EXPANSION
34
t i (ω)
PROJECTION
35
STCF's
SPLINE MODELING & MAPPING
36
FETF MODEL
37
CONVERT TO AUDIO
38
OUTPUT AUDIO
39
DONE
40

FIG. 4

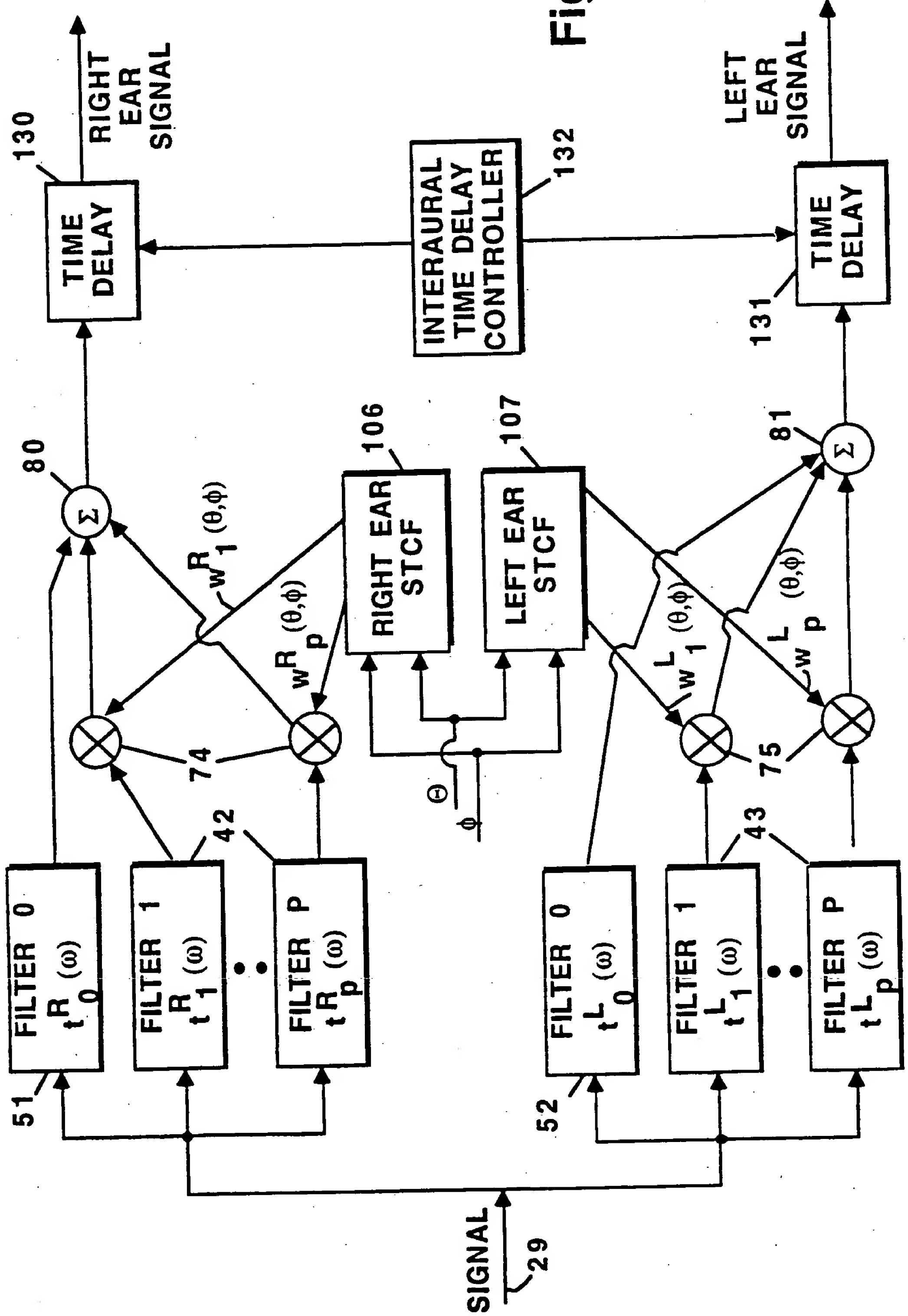

SIGNAL
29
51
52
FILTER 0
FILTER 1
FILTER P
FILTER 0
FILTER 1
FILTER P
42
43
74
75
Θ
φ
RIGHT EAR STCF
LEFT EAR STCF
106
107
80
81
130
131
TIME DELAY
TIME DELAY
INTERAURAL TIME DELAY CONTROLLER
132
RIGHT EAR SIGNAL
LEFT EAR SIGNAL

Fig. 5a

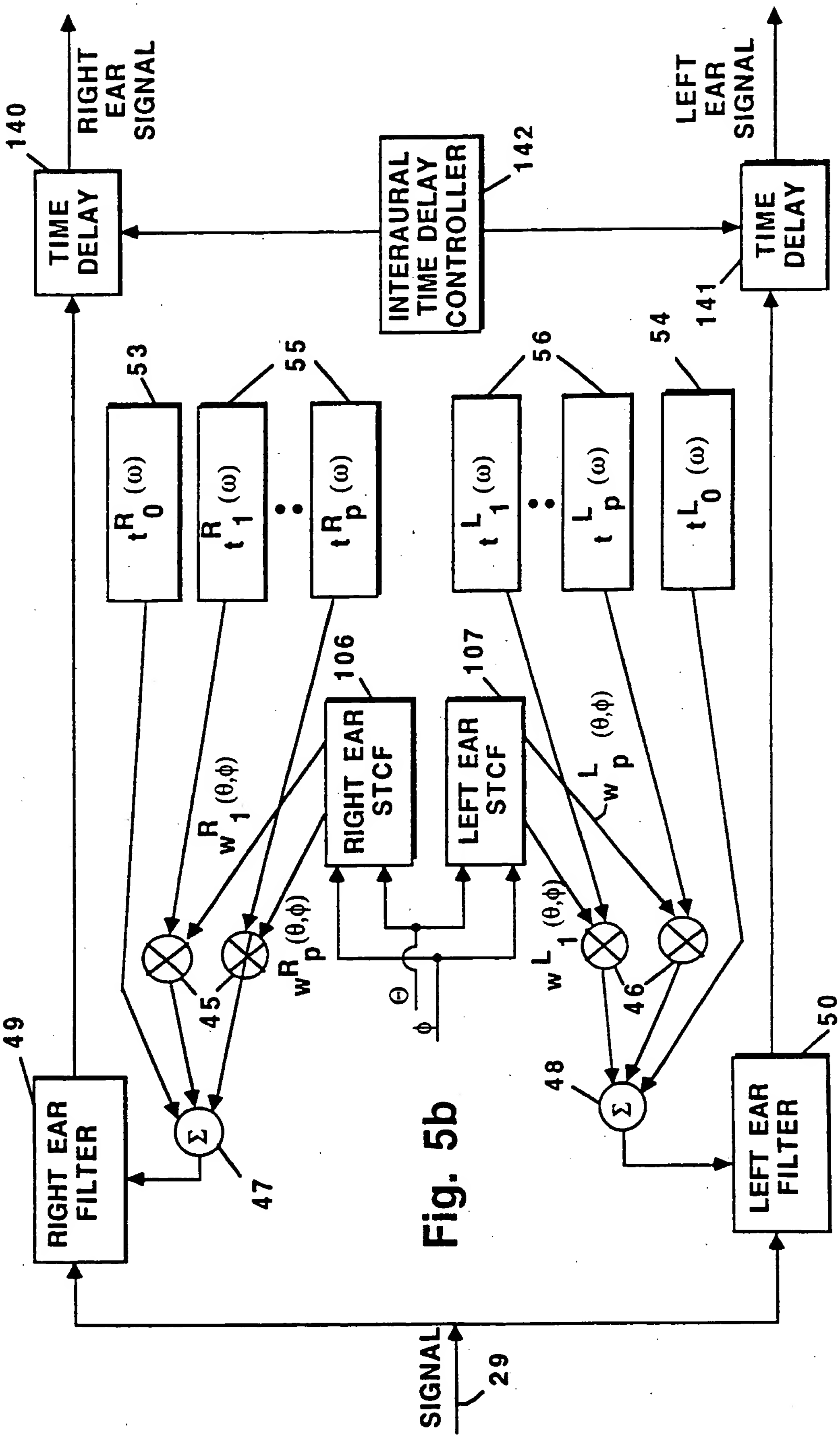
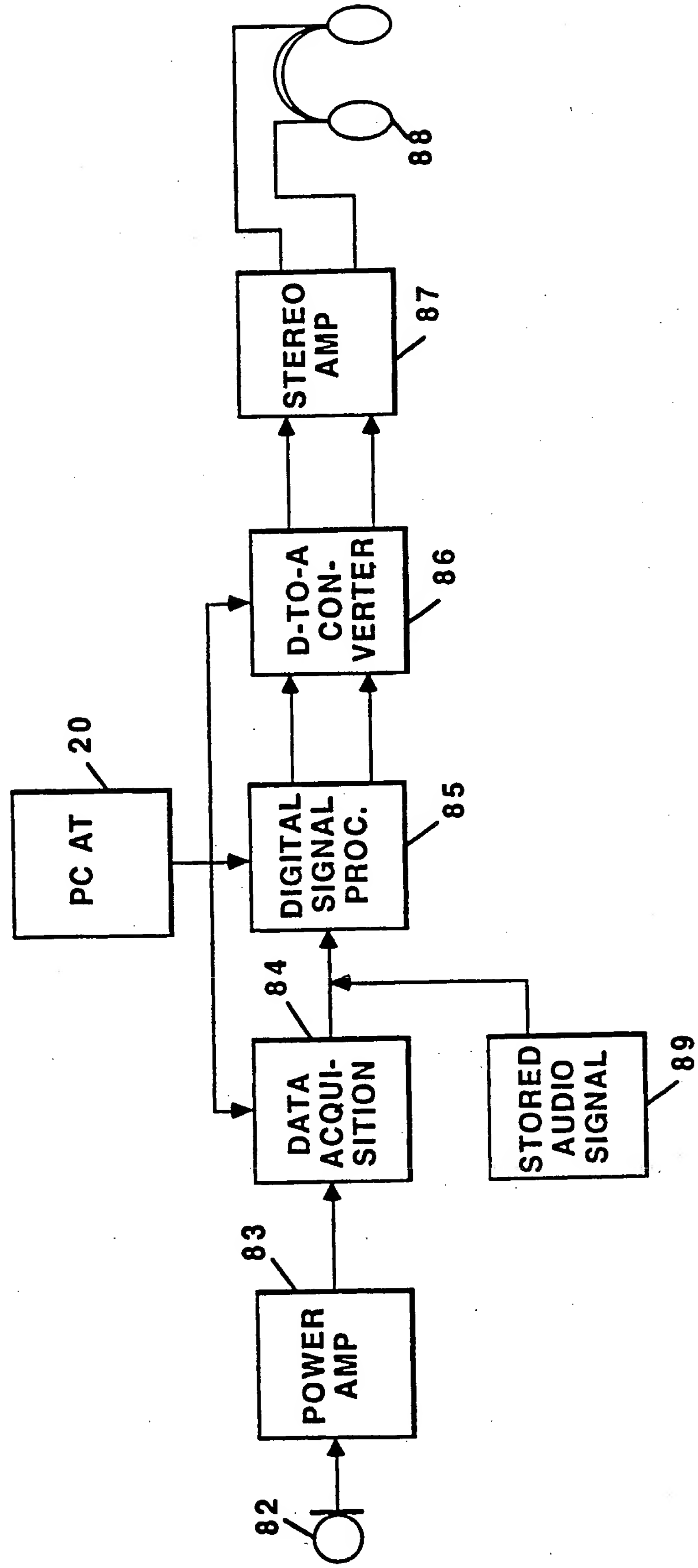

RIGHT EAR SIGNAL
140
TIME DELAY
INTERAURAL TIME DELAY CONTROLLER
142
LEFT EAR SIGNAL
TIME DELAY
141
53
55
56
54
$t^R_0(\omega)$
$t^R_1(\omega)$
$t^R_p(\omega)$
$t^L_1(\omega)$
$t^L_p(\omega)$
$t^L_0(\omega)$
106
107
RIGHT EAR STCF
LEFT EAR STCF
$w^R_1(\theta,\phi)$
$w^R_p(\theta,\phi)$
$w^L_1(\theta,\phi)$
$w^L_p(\theta,\phi)$
45
46
θ
φ
49
RIGHT EAR FILTER
47
48
50
LEFT EAR FILTER
SIGNAL
29

Fig. 5b



88
STEREO AMP
87
D-TO-A CON-VERTER
86
PC AT
20
DIGITAL SIGNAL PROC.
85
84
DATA ACQUI-SITION
STORED AUDIO SIGNAL
89
83
POWER AMP
82

Fig. 6

INTERNATIONAL SEARCH REPORT

International Application No PCT/US 93/01840

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)[6]

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC^5: H 04 R 5/00

II. FIELDS SEARCHED

Minimum Documentation Searched[7]

Classification System	Classification Symbols
IPC^5	H 04 R, H 04 S

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched[8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category [*]	Citation of Document,[11] with indication, where appropriate, of the relevant passages[12]	Relevant to Claim No.[13]
A	DE, A1, 3 934 671 (AKG) 26 April 1990 (26.04.90), the whole document. --	1
A	DE, A1, 3 922 118 (KÖNIG) 17 January 1991 (17.01.91), abstract; column 1, line 1 - column 4, line 45; fig. 1a,1b; claim 1. --	1
A	EP, A1, 0 249 640 (SONY) 23 December 1987 (23.12.87). ----	

* Special categories of cited documents:[10]

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
02 June 1993	16. 06. 93
International Searching Authority	**Signature of Authorized Officer**
EUROPEAN PATENT OFFICE	GRÖSSING e.h.

Form PCT/ISA/210 (second sheet) (January 1985)

ANHANG

zum internationalen Recherchenbericht über die internationale Patentanmeldung Nr.

ANNEX

to the International Search Report to the International Patent Application No.

ANNEXE

au rapport de recherche international relatif à la demande de brevet international n°

PCT/US 93/01840 SAE 71501

In diesem Anhang sind die Mitglieder der Patentfamilien der im obengenannten internationalen Recherchenbericht angeführten Patentdokumente angegeben. Diese Angaben dienen nur zur Unterrichtung und erfolgen ohne Gewähr.

This Annex lists the patent family members relating to the patent documents cited in the above-mentioned international search report. The Office is in no way liable for these particulars which are given merely for the purpose of information.

La présente annexe indique les membres de la famille de brevets relatifs aux documents de brevets cités dans le rapport de recherche international visée ci-dessus. Les reseignements fournis sont donnés à titre indicatif et n'engagent pas la responsibilité de l'Office.

Im Recherchenbericht angeführtes Patentdokument Patent document cited in search report Document de brevet cité dans le rapport de recherche	Datum der Veröffentlichung Publication date Date de publication	Mitglied(er) der Patentfamilie Patent family member(s) Membre(s) de la famille de brevets	Datum der Veröffentlichung Publication date Date de publication
DE A1 3934671	26-04-90	AT A 2635/88	15-11-90
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		GB B2 2224186	25-11-92
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